transmission parameter may be constituted by a lag value corresponding to a pitch period, or alternatively, by a pitch gain value. It is also possible to configure the voice coding method in such a manner that the sample positions of non-zero amplitude values are revised within the region corresponding to the lag value in association with the comparative relation between lag value magnitudes or with the pitch gain value.

According to Ozawa, an <u>adaptive code book</u> is configured to conduct pitch prediction by computing a pitch prediction signal and obtaining a delay that corresponds to a pitch period of the input speech signal. In particular, according to Fig. 1 of Ozawa, the adaptive code book circuit 300 receives the past excitation signal V(n) from the weighting signal calculator 360, the output signal $x_w(n)$ from the subtracter 235 and the perceptually weighted impulse response $K_w(n)$ from the impulse response calculator 310 and determines the value of T at which the distortion D_T of the following equation becomes the minimum as delay T that corresponds to the pitch period.

$$D_T = \sum_{n=0}^{N-1} x'^{2}_{w}(n) - \frac{\left[\sum_{n=0}^{N-1} x'_w(n) y_w(n-T)\right]^2}{\sum_{n=0}^{N-1} y_w^2(n-T)}$$

Here, $y_w(n-T) = v(n-T) * h_w(n)$ represents a pitch prediction signal and the symbol * represents a convolution operation. The gain β is expressed as follows.

$$\beta = \frac{\sum_{n=0}^{N-1} x'_w(n) y_w(n-T)}{\sum_{n=0}^{N-1} y_w^2(n-T)}$$

The adaptive codebook circuit 300 conducts the pitch prediction from the obtained delay and gain by the following equation and output the prediction error signal $z_w(n)$ to the excitation quantizer 350.

$$z_w(n) = x'_w(n) - \beta v(n - T) * h_w(n)$$

In contrast to the method of Ozawa, according to Applicants' claimed invention, configuration variable codebook 1 or 1' shown in Figs. 5 or 6 is an <u>algebraic code book</u> that generates code vectors each comprising a plurality of non-zero sample values, and is further capable of re-configuring itself by changing positions of non-zero samples in accordance to an index *i* and a transmission parameter *p* such as a pitch period (a lag value). Configuration variable codebook 1 or 1' changes in terms of the non-zero sample positions, but importantly and unlike Ozawa <u>does not change in respect to the number of the non-zero samples</u>. As a result of this, the bit number required for transmitting a code vector index does not have to be increased with a change made in terms of the non-zero sample positions. Applicants' coder and decoder are further discussed with reference to Figs. 5 – 8 of Applicants' drawing.

In regard to a coder that is configured in a manner shown in Fig. 5 of the present invention, the configuration variable codebook 1, after adjusting non-zero sample positions based on index *i* and transmission parameter *p*, outputs a code vector C_i, which

is multiplied with a gain g at the gain unit 2. Then the linear prediction synthesis filter 3 receives the result of the multiplication and outputs a reproduced signal gAC_i. The subtracter 4 deducts the reproduced signal gAC_i from the input signal X and outputs the variance that result, which is an error signal E. The error power evaluation unit 5 computes an error power from the error signal E. The above process is performed for all the code vectors C_i that are output from the configuration variable codebook 1 and also for all the varieties of the gain g to determine a pair of values, the index i of the code vector C_i and the gain variety, with which the associated error power is the smallest. Then, determined pairs of these values are transmitted to a decoder.

In regard to a decoder that is configured in a manner shown in Fig. 6, the parameter separation unit 6 obtains respective parameters contained in the data received from the coder. The configuration variable codebook 1' outputs a code vector C_i, based on the index i and the transmission parameter p, among parameters obtained by the parameter separation unit 6. The gain unit 2', then multiplies the gain g, which is another one of the parameters obtained by the parameter separation unit 6 to the code vector Ci. The linear prediction synthesis filter 3' receives the result of multiplication and outputs the reproduction signal gAC_i. Linear prediction parameters, while not shown in Fig. 6, are input to the linear prediction synthesis filter 3' from the parameter separation unit 6.

The configuration variable codebook 11 or 11' in Fig.7 or 8 is constituted from a pair of a non-zero sample position control unit 16, which receives an index i and a transmission parameter, a pitch period (a lag value) l, and a pitch emphasis filter 17,

which receives a signal output from the non-zero sample position control unit 16 and the pitch period (the log value) *1*. The non-zero sample position control unit 16 controls the positions of non-zero samples so that they are changed accordingly to the pitch period (the lag value) *1*, without changing the number of the non-zero samples. In doing this control, the non-zero sample position control unit 16 firstly distributes non-zero samples within an area corresponding to the lag value. At the same time, the non-zero sample position control unit 16 controls the manner of positioning the non-zero samples in which the non-zero samples positioned in a portion of the area corresponding to a lag value larger than the half of the frame length is reduced in number, where an effect of a recursive process performed by the pitch emphasis filter 17 is smaller, in a case in which the length indicated by the lag value is longer than half the length of the frame length. As a result of having this manner of non-zero sample position control, it becomes possible to keep the non-zero samples to a constant number irrespective to changes in the lag value and the frame length and hence avoid a situation in which the bit number required for transmitting a code vector index increases.

As reviewed above, an adaptive codebook is configured to conduct pitch prediction by computing a pitch prediction signal and obtaining a delay that corresponds to a pitch period of the input speech signal. In contrast, a configuration variable codebook is capable of re-configuring itself by changing the non-zero sample positions based on an index i and a transmission parameter p such as a pitch period (a lag value). The configuration variable codebook changes the positions of non-zero samples but does not

change the number of them. As result of this, the bit number required for transmitting a code vector index does not have to be increased.

Ozawa teaches digitizing sounds of a speech without degrading the associated sound-quality and offers a configuration in which an excitation quantizer 350 determines the most optimum set of positions for M non-zero pulses of which the vibration amplitudes are obtained and actually obtains the amplitudes of the M non-zero pulses. Fig.8 of Ozawa shows a block diagram of a speech coding apparatus, which determines and obtains the positions and amplitudes of these non-zero pulses, of which the pulse series, when converted back to a speech signal, gives the minimum deviation against the original speech signal. According to Ozawa, the positions and amplitudes of the non-zero amplitude pulses are determined for each of the received speech signals in a manner with which they are determined to be the most optimum for respective speech signals. In addition, information on the non- zero pulse positions and their amplitudes needs to be quantized for transmission. It, therefore, becomes necessary to transmit certain supplementary information together with the quantized signal to enable decoding the pulse series at a receiver side.

In contrast, the present invention differs as follows. A non-zero amplitude pulse series is generated by the configuration variable codebook, which is determined in accordance to a rule that is set by another transmission parameter such as a pitch period value. The positions and amplitudes are not determined respectively for the received speech signals,

but are selected as the most optimum series from entries in the configuration variable codebook of which the contents are varied and optimized by a certain rule.

As a result of this configuration associated with the present invention, it is not required to transmit any supplementary information. The codebook configuration is varied based on only the correlation existing among data pieces ordinarily transmitted. This means that no bit-rate increase becomes necessary in association with changing positions of non-zero pulses when working with the configuration according to the present invention. An adaptive codebook such as is used by Ozawa is composed of a pulse series each corresponding to the excitation signal in a preceding frame, and thereby may not cause a configuration change in accordance with the methods of the present invention.

These claimed features are not disclosed or suggested by Ozawa. Accordingly, Applicants respectfully submit that claims 1, 7, 13 and 16 stand in condition for allowance. As claims 2 – 6, 8 – 12, 14 – 15 and 17- 18 each depend directly or indirectly from one of allowable claims 1, 7, 13 and 16, Applicants respectfully submit that claims 2 – 6, 8 – 12, 14 – 15 and 17- 18 are allowable for at least this reason.

CONCLUSION

An earnest effort has been made to be fully responsive to the Examiner's objections. In view of the above amendments and remarks, it is believed that claims 1 - 18, consisting of independent claims 1, 7, 13 and 16, and the claims dependent therefrom,

are in condition for allowance. Passage of this case to allowance is earnestly solicited. However, if for any reason the Examiner should consider this application not to be in condition for allowance, he is respectfully requested to telephone the undersigned attorney at the number listed below prior to issuing a further Action.

Attached is a marked up version of the changes made to the specification and claims by the current amendment. The attached pages are captioned **"Version With Markings To Show Changes Made"**.

Any fee due with this paper may be charged on Deposit Account 50-1290.

Respectfully submitted,

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IN THE CLAIMS

Please amend claims 1, 2, 4, 7,8, 10 and 13 – 18 as follows:

1. (Twice Amended) A voice coding method based on analysis-by-synthesis vector quantization comprising:

using a configuration variable code book containing a voice source code vector having only a plurality of non-zero amplitude values; and

variably [controlling] replacing a position of a sample of the non-zero amplitude value in the configuration variable code book using an index and a transmission parameter indicating a feature amount of voice.

2. (Twice Amended) The method according to claim 1, further comprising :

variably [controlling] replacing the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

3. (Unchanged) The method according to claim 2, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

4. (Twice Amended) The method according to claim 1, further comprising:

variably [controlling] <u>replacing</u> the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

5. (Unchanged) The method according to claim 4, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

6. (Unchanged) The method according to claim 5, further comprising:

reconstructing the position of the sample the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on the pitch gain value.

7. (Twice Amended) A voice decoding method for decoding a voice signal coded by a voice coding method based on analysis-by-synthesis vector quantization comprising:

using a configuration variable code book containing a voice source code vector having only a plurality of non-zero amplitude values; and

variably [controlling] <u>replacing</u> a position of a sample of the non- zero amplitude value in the configuration variable code book using an index and a transmission parameter indicating a feature amount of voice.

8. (Twice Amended) The method according to claim 7, further comprising:

variably [controlling] <u>replacing</u> the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

9. (Unchanged) The method according to claim 8, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a ceding unit of the voice.

10. (Twice Amended) The method according to claim 7, further comprising:

variably [controlling] replacing the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

11. (Unchanged) The method according to claim 10, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

12. (Unchanged) The method according to claim 11, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on the pitch gain value.

13. (Twice Amended) A voice coding apparatus based on analysis-by- synthesis vector quantization comprising:

a configuration variable code book unit containing a voice source code vector having only a plurality non-zero amplitude values, wherein

said configuration variable code book unit variably [controls] replaces a position of a sample of the non-zero amplitude value in said configuration variable code book unit using an index and a transmission parameter indicating a feature amount of voice.

14. (Twice Amended) The apparatus according to claim 13, wherein:

said configuration variable code book unit variably [controls] replaces the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

15. (Twice Amended) The apparatus according to claim 13, wherein:

said configuration variable code book unit variably [controls] replaces the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

16. (Twice Amended) A voice decoding apparatus for decoding a voice signal coded by a voice coding apparatus based on analysis-by-synthesis vector quantization comprising:

a configuration variable code book unit containing a voice source code vector having only a plurality of non-zero amplitude values, wherein

said configuration variable code book unit variably [controls] <u>replaces</u> a position of a sample of the non-zero amplitude value using an index and a transmission parameter indicating a feature amount of voice.

17. (Twice Amended) The apparatus according to claim 16, wherein:

said configuration variable code book unit variably [controls] <u>replaces</u> the position of the sample of the non-hero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

18. (Twice Amended) The apparatus according to claim 16, wherein:

said configuration variable code book unit variably [controls] <u>replaces</u> the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.